SIP Forum Update for IETF (SFSIW-2)

Eric Burger
Chairman of the Board
SIP Forum

Who is the SIP Forum

- You: 5,000 Participant Members
- Founded in 2000
- Leading Non-Profit IP Communications Industry Association
- Membership comprised of Full Members that pay annual dues, Academic and Research Institutions, and Individual Participant Members

SIP Forum Full Member Companies

As of 11-17-08

3Com
Possible made practical

Acme Packet

Alcatel·Lucent

Aspect

AudioCodes

AVAYA

bandwidth.com
Complete business communications

BroadSoft
Innovation calling.

CableLabs

CBEYOND

Cedar Point Communications
Simplifying The Network

CGI

Cisco

covergence

COX

Data Connection

Dialogic
Making Innovation Thrive

digium

ERICSSON

hp

IBM

inGate

Microsoft

MITEL

NEC

net
Network Equipment Technologies

NEUSTAR

NOKIA
Connecting People

ORACLE

PAETEC

POLYCOM

RADVISION
the VoIP experts

SAMSUNG

ShoreTel

SIEMENS

snom
VoIP phones

SOLINET
Solutions for Innovative Networks!

Sonus Networks

TANDBERG

TECNOMEN

TEKELEC

Telsis

iol

SIP FORUM

SIPconnect

SIPit

SIP Forum Mission

- Advance the development and deployment of innovative IP communications solutions that comply with, and properly interoperate with, other products and services that use the Session Initiation Protocol (SIP) protocol.

Why Do People Say SIP is Complex?

RFC-ED HOME | NEWS | RFC DATABASE | RFC SEARCH | RFC ERRATA | I-D SEARCH | IETF HOME

RFC Index Search Engine

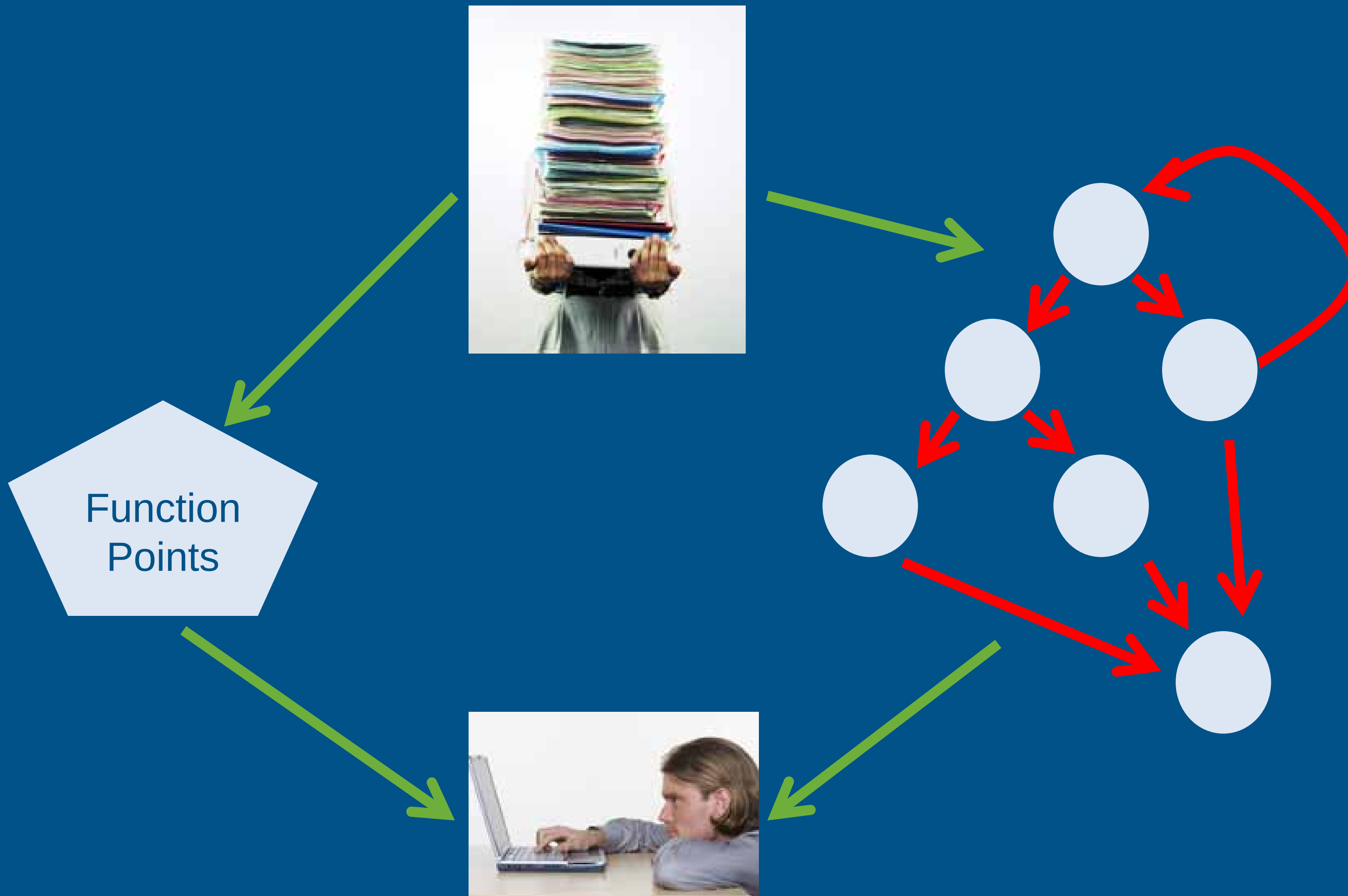
HELP

o Based on your search of [*sip*] in the *All Fields* field **130** matches were found

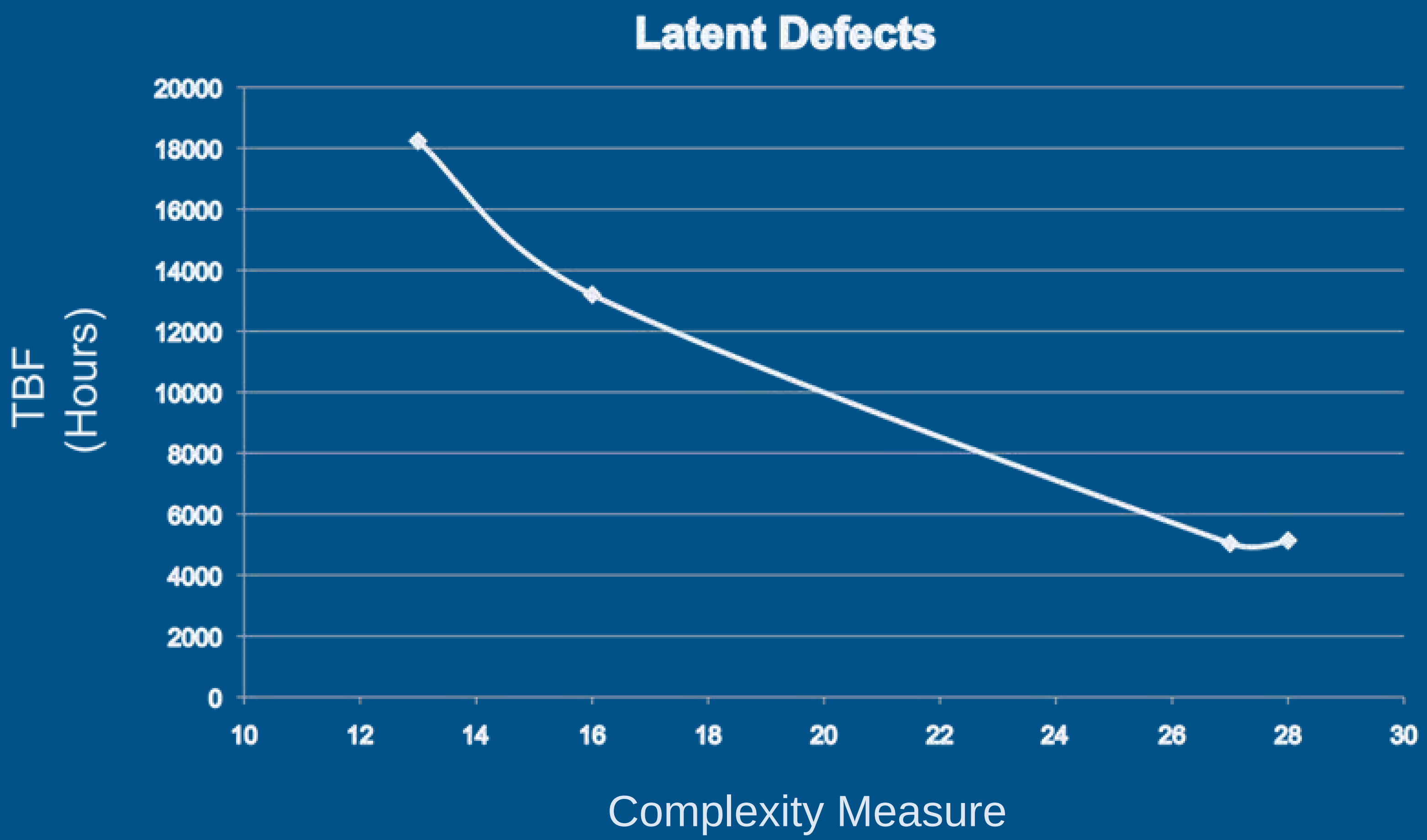
- Below you will find matching items ***1 through 130***

Number	Title	Author or Ed.	Date	Format	More Info (Obs&Upd)	Status
RFC5278	**IANA Registration of Enumservices for Voice and Video Messaging**	J. Livingood, D. Troshynski	July 2008	ASCII		PROPOSED STANDARD
RFC5263	**Session Initiation Protocol (SIP) Extension for Partial Notification of Presence Information**	M. Lonnfors, J. Costa-Requena, E. Leppanen, H. Khartabil	September 2008	ASCII		PROPOSED STANDARD
RFC5196	**Session Initiation Protocol (SIP) User Agent Capability Extension to Presence Information Data Format (PIDF)**	M. Lonnfors, K. Kiss	September 2008	ASCII		PROPOSED STANDARD
RFC5194	**Framework for Real-Time Text over IP Using the Session Initiation Protocol (SIP)**	A. van Wijk, Ed., G. Gybels, Ed.	June 2008	ASCII		INFORMATIONAL
RFC5118	**Session Initiation Protocol (SIP) Torture Test Messages for Internet Protocol**	V. Gurbani, C. Boulton, R. Sparks	February 2008	ASCII	Errata	INFORMATIONAL

Computer Science Definition of Complex

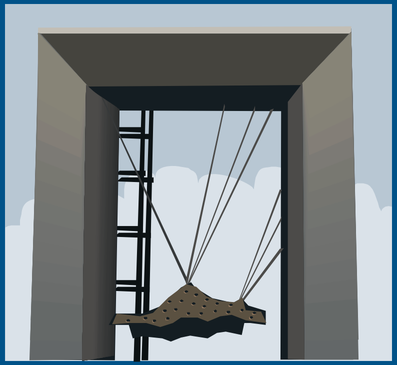


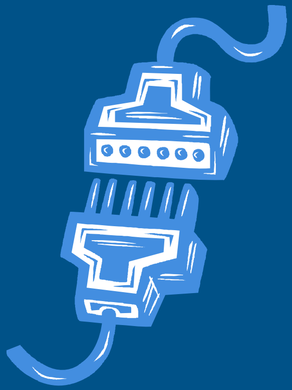
Why Complexity Matters



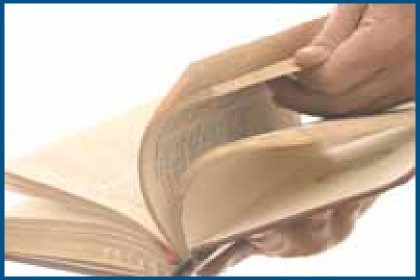
Is it Possible to Reduce Complexity?

What are the Different Kinds of Profiles?





SIPconnect: A Simplifying Profile



Profile Refrain

- Describes what has to be implemented from IETF, W3C, ITU, etc.
  - Anything else not **required** or **expected**
  - **Cannot go to less than base specs** (unspecified stuff won't crash implementations)
  - Most common simplification is making **optional feature mandatory**
- ***Is not protocol***
  - **Everything**, even items mandatory in spec, **negotiated per SIP**
  - Some things left as **local configuration** or **policy** specified by profile
  - **NEVER** "I am a SIPconnect PBX, so you just know my capabilities"

SIP FORUM

SIPconnect Status

Richard Shockey
SIP Forum Technical Work Group Chair
Spencer Dawkins
SIPconnect 1.1 Editor

SIPit

SIPconnect

The Evolution of Enterprise VoIP

- First : Replace the RJ-11
  - Immediate gains in CAPEX as single wiring harness simplifies campus management.
  - Greenfield ROI – NO Brainer
- Second : Replace the TIE Lines
  - Integrate Enterprise wide Dial Plan Management into single IP Network. Immediate OPEX gains.
- Third : Replace the PRI (Today)
  - All IP E2E
- Fourth : Peer with Business Partners
  - The 40-40-20 rule
- Fifth : Seamless Campus/Mobility Integration
  - Its not fixed Mobile Convergence its F/MSubstitution

What is SIPConnect?

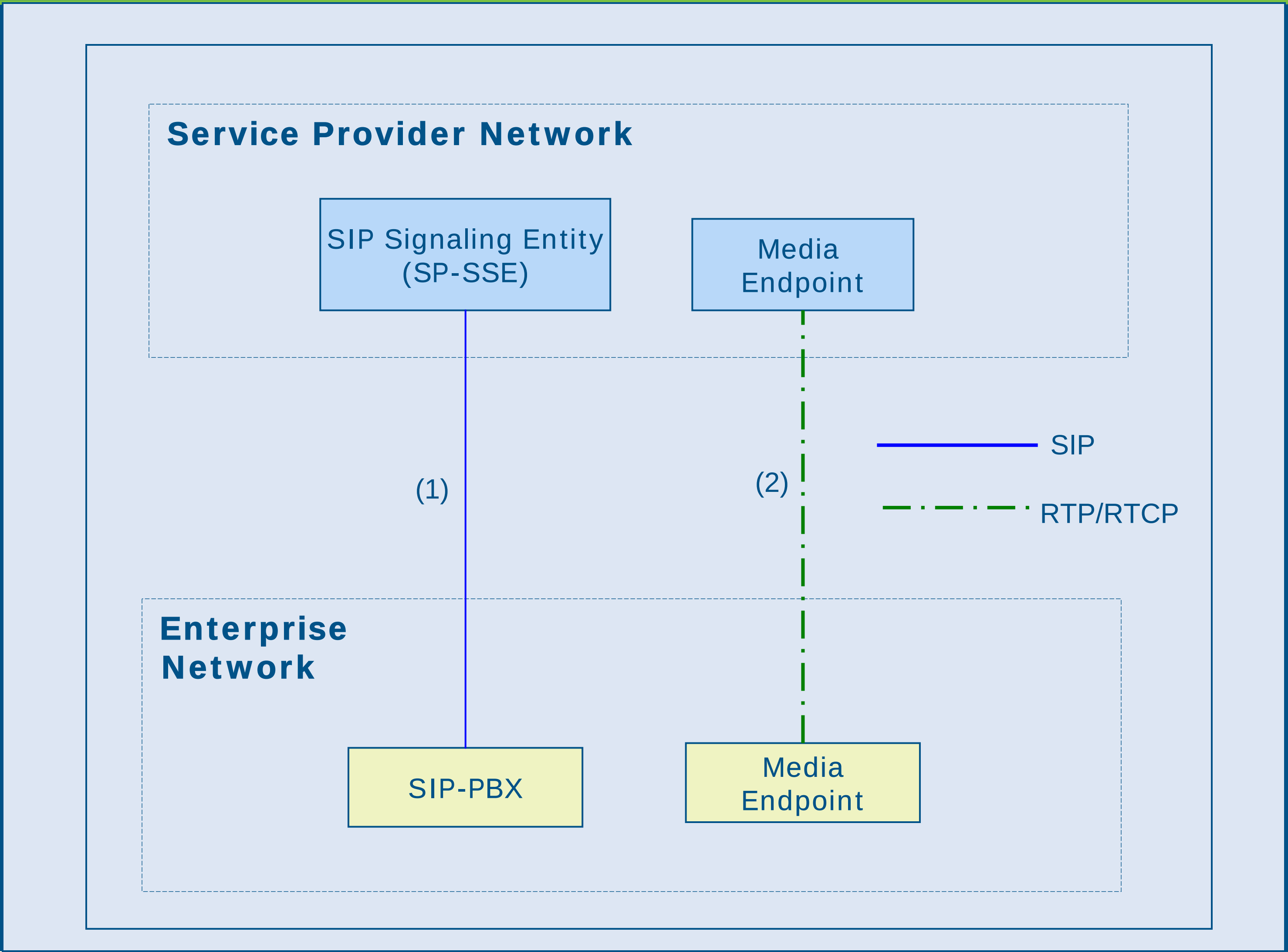
- “ The SIPconnect Technical Recommendation is an important industry initiative that builds on existing IETF standards to define a method for interconnection between IP PBXs and VoIP service provider networks, and specifies a reference architecture, required protocols and features, and implementation rules necessary for seamless IP peering between IP PBXs and VoIP service providers.”

Impact of SIPconnect

- SIP Trunking is beginning to enjoy broad industry adoption, especially among competitive service providers and PBX providers.
- Channel & Industry awareness is strong and growing
- SIPconnect is the de facto standard for SIP Trunking– there are no competitors

By any objective measure SIPconnect has been a positive influence on SIP Trunking adoption and interoperability

SIPconnect Architecture Diagram



Building on Success

- SIPconnect 1.1 in process
  - http://www.sipforum.org/sipconect
  - Building on the success of 1.0
  - *Builds on existing IETF standards*
  - ***Does not change IETF standards***

Current Issues

- 1.0 was “dial tone”
- 1.1 still voice centric but add features functionality and updates relevant RFC etc
- Increase MUST requirements
- 1.1 still work in progress (09)
  - Registration

Fax Interoperability – SFFIW-1

Fax Problem Statement

- Inability to reliably send fax over IP
  - Severely impacts acceptance of VoIP
  - Severely impacts acceptance of SIP
- Hard to determine when to go T.38
  - Many conformant ways of signaling
  - Few do it the same way
- T.30 over G.711 is problematic

1st SIP Forum Fax Interoperability Workshop

- Held in San Francisco, 10 November 2008
- 19 participants
  - SIP-based Fax-over-IP and Voice-over-IP manufacturers, service providers, technologists
- Discussion on where the problems are
  - Transport, ES-AS and AS-AS borders, T.38 interoperability, signaling
- Consensus on problems to attack and commitment to do work

Next Steps for Fax

- Step #1: Publish problem statement
- Step #2: Definitions of successful fax transmission
  - Performance
  - Network topology
  - Network performance
  - Identify parameters in which T.38 works well
- Step #3: Can we use signaling to work around problems?
  - Create checklist including FEC, ECM, UDP redundancy/depth
- Step #4: Marketing Issues
  - Data vs. Voice (T.38 vs. G.711) Whitepaper
  - “Fax Safe” Certification Program
  - Interoperability Reference/Best Practices Document
- Technical Issues
  - Fixing Protocols?
  - New Protocols?
  - These get passed to relevant standards bodies

User Agent Configuration

John Elwell

Chair, SIP-UA Configuration Task Group

Background

- Formed as a result of SFSIW follow-up meeting during IETF 72
- Need to address SIP UA configuration as a barrier to successful deployment of SIP
- Working method: mailing list and conference calls
- In conjunction with SIPit
- Mailing list subscription:
  - http://sipforum.org/mailman/listinfo/ua-config
- About 10 active participants
- Chair John Elwell

Aims

- Simple boot mechanism suitable for a variety of UA types
  - Rich UI soft clients
  - UI-less phones and adaptors
  - Phones with simple UI
  - Enterprise phones covered, but only basic configuration – full enterprise features outside scope
- Small set of essential parameters
  - Proprietary datasets can be downloaded to cover full enterprise features, for example
- Profile of IETF SIP configuration framework

Status

- Aim to identify minimum parameter set and outline/scope for a Technical Recommendation shortly
- Editor required
- Need a reasonably complete proposal for Feb/March 2009 – good enough for implementers to work from
- Testing at SIPit May 2009
- Meeting during breakfast Thursday in Ramsey room

SIP Interoperability Testing – SIPit

Robert Sparks

SIP Forum Board Member

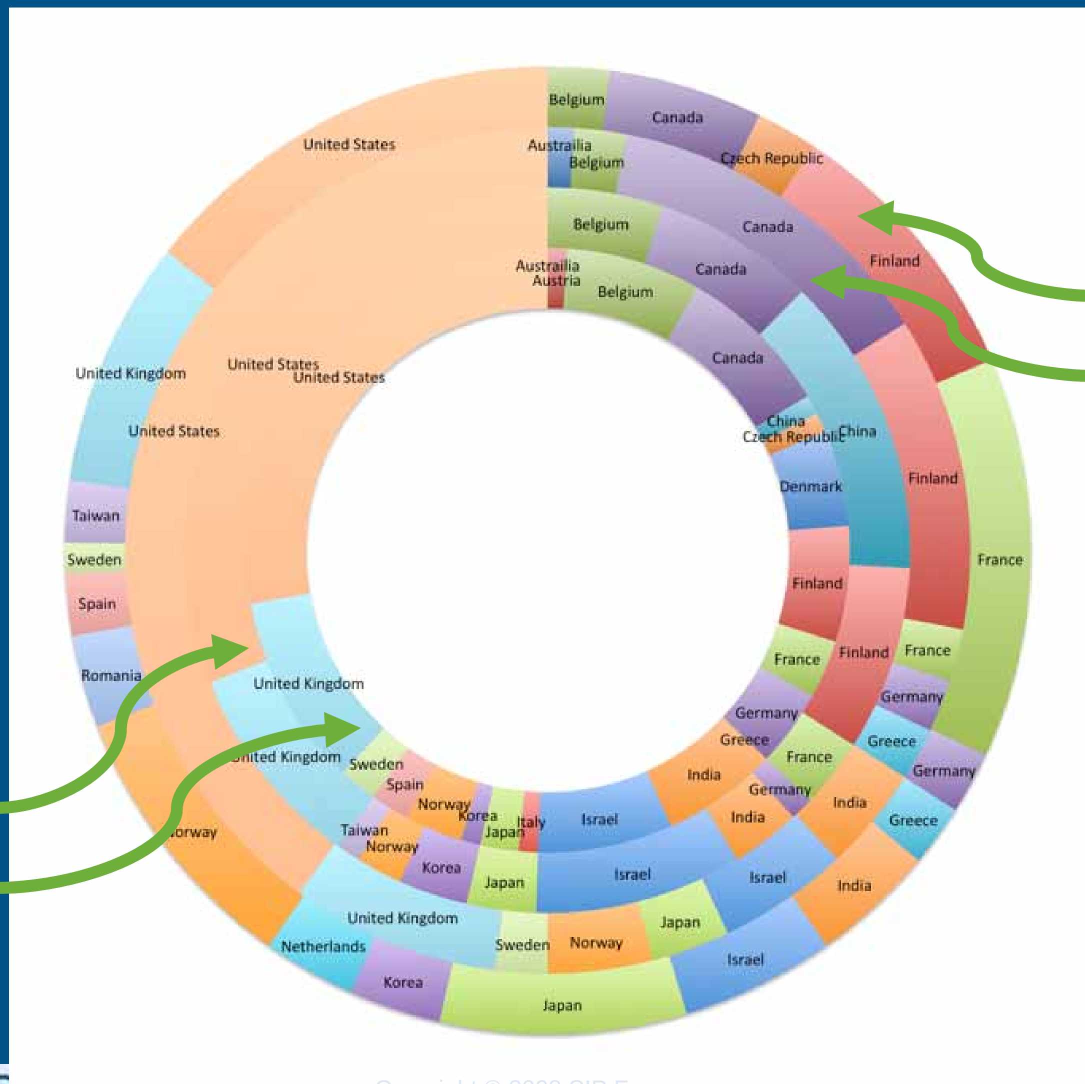
Chair, SIP Forum Test Work Group

SIPit

- Week-long engineering test events
  - Held twice a year
  - Moves around the globe
- Averages around 100 implementations
  - 60 to 90 companies
  - 16 to 20 countries

SIPit

SIPit is an International Event



SIPit 23: France

SIPit 22: US

SIPit 21: China

SIPit 20: Belgium

Goal: Find and fix what doesn't work

- Issues are corrected in real-time
  - Allows immediate retesting
- Very high-yield testing
  - Participants claim 4 to 6 months of results from the week at SIPit compared to what they would achieve testing separately

Goal: Find and fix what doesn't work

- The Standards improve
  - If two teams have to argue about the specification, the specification needs to be corrected
  - Testing identifies errors and omissions in the Standards
  - Reports from the SIPit allow the IETF to remain focused on real industry needs

Past Events

Bakeoff

SIPit 1 (April 1999) - SIPit 23 (October 2008)

Prior Hosts

Upcoming Events

- SIPit 24 – May 2009 – Tokyo, Japan
  - Hosted by JPNIC and NICT

JPNIC

NICT National Institute of Information and Communications Technology

More Information

SIP it

SIP FORUM

www.sipit.net

SIP Forum Future Work – IETF Community Input

Expanding Focus Beyond Voice

Arnoud van Wijk
Real Time Text Taskforce
Participant Member, SIP Forum

Real-Time Text Taskforce

The SIP Forum is an industry association with members from the leading IP communications companies.

*Its **mission: To advance the adoption of IP communications products and services based on SIP**.*
*The Forum promotes SIP as the technology of choice for the **control of real-time multimedia communication sessions** throughout the Internet, corporate networks, and wireless networks.*

Yet SIPConnect 1.1 : " The primary service to be delivered over this interface is **audio-based** call origination and/or termination between the Enterprise and Service Provider network. The delivery of any other service (e.g. video-based services, instant messaging, etc.) is out of scope.

Saying that service providers **MUST** accommodate unsupported (or even unrecognized) media types by passing the SDP media lines unchanged, and **SHOULD** pass unsupported (or even unrecognized) SIP headers and bodies **is not sufficient for making support for other media possible.**

Real-Time Text Taskforce

*Real-Time Text (Text over IP / RFC4103) and video are real-time CONVERSATIONAL media (**voice equivalent communication**). And should be considered BASIC media that must always be supported and possible on SIP networks. (different then IM, etc)*

ECRIT: emergency calls containing audio, video and real-time text must be supported according to RFC 5012 and draft-ietf-ecrit-phonebcp

3GPP/NGN already supports video and real-time text via the IMS

UN convention on the rights for people with disabilities:
Article 21 - Freedom of expression and opinion, and access to information
States Parties shall take all appropriate measures to ensure that persons with disabilities can exercise the right to freedom of expression and opinion, including the freedom to **seek, receive and impart information and ideas on an equal basis with others and through all forms of communication of their choice,** as defined in article 2 of the present Convention, including by:
b) Accepting and facilitating the use of **sign languages,** Braille, augmentative and **alternative communication, and all other accessible means, modes and formats of communication** of their choice by persons with disabilities in official interactions;

www.realtimetext.org

Real-Time Text Taskforce

The longer we delay adding real-time text and video as basic media, the harder it will be to have those media as normal modes of communication. And to be able to use them whenever.

<u>Now is the moment to add real-time text and video, else we are voice only again.</u>

Question to SIPFORUM:

How do we bring real-time text and video back as basic media, making it just as available as voice is.

(bandwidth considerations on SIP VoIP networks would only be relevant for video, NOT for real-time text/ToIP)

Thank you for listening!

Contact: arnoud@realtimetext.org

Topics from SFSIW-1 (IETF 70) [1/2]

- Highlight that UAs must support INVITEs and re-INVITEs without offers.
- Create a number of small BCPs that highlight the right way to implement SIP features, and that interoperability requires the implementation of these features.
- Create a few, big BCPs that describe which features a UA must support for interoperability and the right way to implement those features.
- Publish a BCP indicating minimum field lengths a UA needs to accept in a SIP header.

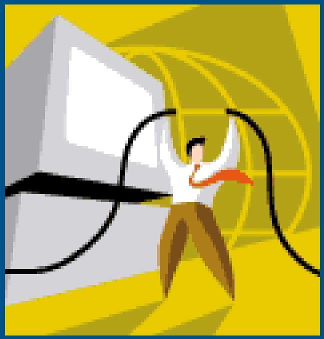
Topics from SFSIW-1 (IETF 70) [2/2]

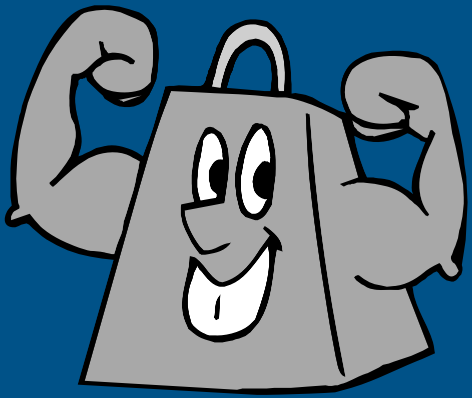
- Create a (set of) profile(s) that guarantee interoperability.
- Create a “SIP HD Video” profile and certification program.
- Write test cases around basic, interoperable features.
- Identify reference endpoints that have full feature implementation and have them available for face-to-face (e.g., at SIPit) and remote testing.
- Encourage vendors to publish interoperability reports in a standard format

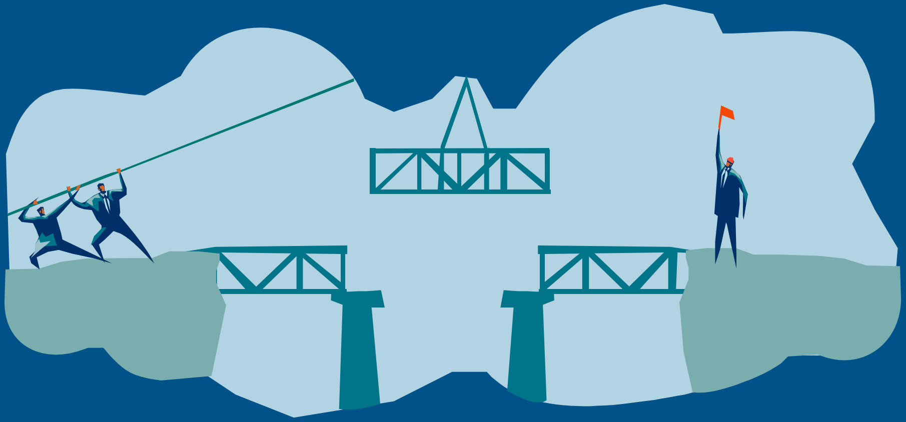
Discussion

- Other topics?
- Renew focus on particular SFSIW-1 topics?
- Finish what we have on docket before we take on too much?

SIP is Strong and Getting Stronger







Thank You For Participating

http://www.sipforum.org